

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
REQUEST FOR FILING NATIONAL PHASE OF
PCT APPLICATION UNDER 35 U.S.C. 371 AND 37 CFR 1.494 OR 1.495

09/830028

To: Hon. Commissioner of Patents
 Washington, D.C. 20231



00909

TRANSMITTAL LETTER TO THE UNITED STATES
 DESIGNATED/ELECTED OFFICE (DO/EO/US)

Atty Dkt: P 279295 /2980530US/PG/KP
M# /Client Ref.

From: Pillsbury Winthrop LLP, IP Group:

Date: April 20, 2001

This is a **REQUEST** for **FILING** a PCT/USA National Phase Application based on:

1. International Application	2. International Filing Date	3. Earliest Priority Date Claimed
<u>PCT/FI99/00868</u>	<u>19</u> <u>October</u> <u>1999</u>	<u>21</u> <u>October</u> <u>1998</u>
<u>FI</u> country code	Day MONTH Year	Day MONTH Year (use item 2 if no earlier priority)

4. Measured from the earliest priority date in item 3, this PCT/USA National Phase Application Request is being filed within:

(a) ☐ 20 months from above item 3 date (b) ☒ 30 months from above item 3 date,

(c) Therefore, the due date (unextendable) is April 21, 2001

5. Title of Invention DIGITAL TELECOMMUNICATION SYSTEM

Inventor(s) VERKAMA, Markku

Applicant herewith submits the following under 35 U.S.C. 371 to effect filing:

7. ☒ Please immediately start national examination procedures (35 U.S.C. 371 (f)).

8. ☐ A copy of the International Application as filed (35 U.S.C. 371(c)(2)) is transmitted herewith (file if in English but, if in foreign language, file only if not transmitted to PTO by the International Bureau) including:

- a. ☐ Request;
 b. ☐ Abstract;
 c. pgs. Spec. and Claims;
 d. sheet(s) Drawing which are ☐ informal ☐ formal of size ☐ A4 ☐ 11"

9. ☒ A copy of the International Application has been transmitted by the International Bureau.

10. A translation of the International Application into English (35 U.S.C. 371(c)(2))

- a. ☒ is transmitted herewith including: (1) ☒ Request; (2) ☒ Abstract;
 (3) 16 pgs. Spec. and Claims;
 (4) 2 sheet(s) Drawing which are:
 ☐ informal ☒ formal of size ☒ A4 ☐ 11"
- b. ☐ is not required, as the application was filed in English.
 c. ☐ is not herewith, but will be filed when required by the forthcoming PTO Missing Requirements Notice per Rule 494(c) if box 4(a) is X'd or Rule 495(c) if box 4(b) is X'd.
 d. ☐ Translation verification attached (not required now).

11. ☐ Attached:
12. ☐ Preliminary Amendment:
13. ☒ Basic U.S. National fee per Rule 492(a)(1)-(4) was previously timely filed.:
14. **Calculation of remaining fees due (if any):** based on amended claim(s) per above item
☐ 12 (above) or item(s) in PAT-112 (filed previously) ☐ 12 ☐ 14 ☐ 17 ☐ 25
15. **CLAIMS FEES** ☒ previously paid ☐ paid herewith as follows:
- 15A. Small Entity Statement ☐ Herewith ☐ Previously Filed

				Large/Small Entity		Fee Code
16. Total Effective Claims	17	minus 20 =	0	x \$18/\$9	+0	966/967
17. Independent Claims	2	minus 3 =	0	x \$80/\$40	+0	964/965
18. If <u>any proper</u> multiple dependent claim (ignore improper) is present,				\$270/\$135	+0	968/969
19. Filing Declaration late, fee paid <input type="checkbox"/> previously <input checked="" type="checkbox"/> now				\$130/\$65	+130	154/254
20. SUBTOTAL					\$130	
21. Original due date: July 22, 2001						
22. Petition is hereby made to extend the <u>original</u> due date to cover the date this response is filed for which the requisite fee is attached				(1 mo) \$110/\$55 =	+110	115/215
				(2mos) \$390/\$195 =		116/216
				(3mos) \$890/\$445 =		117/217
				(4mos) \$1390/\$695 =		118/218
23. If "non-English" box 3 is X'd, add Rule 17(k) processing fee				\$130	+0	156
24. If "assignment" box 6 is X'd, add recording fee.				\$40	+40	581
25. TOTAL FEE ENCLOSED =					\$280	

(Our Deposit Account No. 03-3975)
 (Our Order No. 60258 | 279295
 C# M#

CHARGE STATEMENT: The Commissioner is hereby authorized to charge any fee specifically authorized hereafter, or any missing or insufficient fee(s) filed, or asserted to be filed, or which should have been filed herewith or concerning any paper filed hereafter, and which may be required under Rules 16-18 (missing or insufficient fee only) now or hereafter relative to this application and the resulting Official document under Rule 20, or credit any overpayment, to our Account/Order Nos. shown above for which purpose a duplicate copy of this sheet is attached.
 This CHARGE STATEMENT does not authorize charge of the issue fee until/unless an issue fee transmittal form is filed.

**Pillsbury Winthrop LLP
 Intellectual Property Group**

1600 Tysons Boulevard
 McLean, VA 22102
 Tel: (703) 905-2000
 CHM/JRH

By Atty: Christine H. McCarthy Reg. No. 41844
 Sig: [Signature] Fax: (703) 905-2500
 Tel: (703) 905-2143

NOTE: File in duplicate with PTO receipt (PAT-103A) and attachments

RE: USA National Phase Filing of PCT /FI99/00868

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11. ☒ Please see the attached Preliminary Amendment
12. ☐ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371(c)(3)), i.e., before 18th month from first priority date above in item 3, are transmitted herewith (file only if in English) including:
13. ☒ PCT Article 19 claim amendments (if any) have been transmitted by the International Bureau
14. ☐ Translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)), i.e., of claim amendments made before 18th month, is attached (required by 20th month from the date in item 3 if box 4(a) above is X'd, or 30th month if box 4(b) is X'd, or else amendments will be considered canceled).
15. **A declaration of the inventor** (35 U.S.C. 371(c)(4))
 a. ☐ is submitted herewith ☐ Original ☐ Facsimile/Copy
 b. ☒ is not herewith, but will be filed when required by the forthcoming PTO Missing Requirements Notice per Rule 494(c) if box 4(a) is X'd or Rule 495(c) if box 4(b) is X'd.
16. **An International Search Report (ISR):**
 a. Was prepared by ☐ European Patent Office ☐ Japanese Patent Office ☒ Other
 b. ☒ has been transmitted by the international Bureau to PTO.
 c. ☒ copy herewith (2 pg(s).) ☒ plus Annex of family members (1 pg(s).).
- International Preliminary Examination Report (IPER):**
 a. ☒ has been transmitted (if this letter is filed after 28 months from date in item 3) in English by the International Bureau with Annexes (if any) in original language.
 b. ☒ copy herewith in English.
 c.1 ☒ IPER Annex(es) in original language ("Annexes" are amendments made to claims/spec/drawings during Examination) including attached amended:
 c.2 ☒ Specification/claim pages #14, 15 & 16 claims # 1 - 14
 Dwg Sheets #
 d. ☒ Translation of Annex(es) to IPER (required by 30th month due date, or else annexed amendments will be considered canceled).
18. **Information Disclosure Statement** including:
 a. ☒ Attached Form PTO-1449 listing documents
 b. ☐ Attached copies of documents listed on Form PTO-1449
 c. ☒ A concise explanation of relevance of ISR references is given in the ISR.
19. ☐ **Assignment** document and Cover Sheet for recording are attached. Please mail the recorded assignment document back to the person whose signature, name and address appear at the end of this letter.
20. ☐ Copy of Power to IA agent.
21. ☐ **Drawings** (complete only if 8d or 10a(4) not completed): sheet(s) per set: ☐ 1 set informal; ☐ Formal of size ☐ A4 ☐ 11"
22. Small Entity Status ☒ is **Not** claimed ☐ is claimed (pre-filing confirmation required)
 22(a) (No.) Small Entity Statement(s) enclosed (since 9/8/00 Small Entity Statements(s) not essential to make claim)
23. **Priority** is hereby claimed under 35 U.S.C. 119/365 based on the priority claim and the certified copy, both filed in the International Application during the international stage based on the filing in (country) FINLAND of:
- | Application No. | | Filing Date | Application No. | | Filing Date |
|-----------------|--------|------------------|-----------------|--|-------------|
| (1) | 982283 | October 21, 1998 | (2) | | |
| (3) | | | (4) | | |
| (5) | | | (6) | | |
- a. ☒ See Form PCT/IB/304 sent to US/DO with copy of priority documents. If copy has not been received, please proceed promptly to obtain same from the IB.
 b. ☐ Copy of Form PCT/IB/304 attached.

SCANNED # 224

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24. Attached:

25 Per Item 17.c2, **cancel original** pages #__, claims #__, Drawing Sheets #26. **Calculation of the U.S. National Fee (35 U.S.C. 371 (c)(1)) and other fees is as follows:**Based on amended claim(s) per above item(s) ☐ 12, ☐ 14, ☐ 17, ☐ 25 (hilitte)

Total Effective Claims	minus 20 =	x \$18/\$9	= \$0	966/967
Independent Claims	minus 3 =	x \$80/\$40	= \$0	964/965
If any proper (ignore improper) Multiple Dependent claim is present,		add \$270/\$135	+0	968/969

BASIC NATIONAL FEE (37 CFR 1.492(a)(1)-(4)): →→ **BASIC FEE REQUIRED, NOW** →→→→A. If country code letters in item 1 are not "US", "BR", "BB", "TT", "MX", "IL", "NZ", "IN" or "ZA"

See item 16 re:

1. Search Report was <u>not</u> prepared by EPO or JPO -----	add \$1000/\$500	960/961
2. Search Report was prepared by EPO or JPO -----	add \$860/\$430 +1000	970/971

SKIP B, C, D AND E UNLESS country code letters in item 1 are "US", "BR", "BB", "TT", "MX", "IL", "NZ", "IN" or "ZA"

→ <input type="checkbox"/> B. If <u>USPTO</u> did not issue <u>both</u> International Search Report (ISR) <u>and</u> (if box 4(b) above is X'd) the International Examination Report (IPER), -----	add \$1000/\$500	+0	960/961
(only) → <input type="checkbox"/> C. If <u>USPTO</u> issued ISR but not IPER (or box 4(a) above is X'd), -----	add \$710/\$355	+0	958/959
(one) → <input type="checkbox"/> D. If <u>USPTO</u> issued IPER but IPER Sec. V boxes <u>not all</u> 3 YES, -----	add \$690/\$345	+0	956/957
(these) (4) → <input type="checkbox"/> E. If international preliminary examination fee was paid to <u>USPTO</u> and Rules 492(a)(4) and 496(b) <u>satisfied</u> (IPER Sec. V <u>all</u> 3 boxes YES for <u>all</u> claims), -----	add \$100/\$50	+0	962/963

27. **SUBTOTAL = \$1000**

28. If Assignment box 19 above is X'd, add Assignment Recording fee of ---\$40 +0 (581)

29. Attached is a check to cover the ----- **TOTAL FEES \$1000**

Our Deposit Account No. 03-3975

Our Order No. 60258 | 279295
C# M#

00909

CHARGE STATEMENT: The Commissioner is hereby authorized to charge any fee specifically authorized hereafter, or any missing or insufficient fee(s) filed, or asserted to be filed, or which should have been filed herewith or concerning any paper filed hereafter, and which may be required under Rules 16-18 and 492 (missing or insufficient fee only) now or hereafter relative to this application and the resulting Official document under Rule 20, or credit any overpayment, to our Account/Order Nos. shown above for which purpose a duplicate copy of this sheet is attached.**This CHARGE STATEMENT does not authorize charge of the issue fee until/unless an issue fee transmittal form is filed****Pillsbury Winthrop LLP
Intellectual Property Group**By Atty: Christine H. McCarthyReg. No. 41844Sig: Fax: (202) 822-0944Tel: (202) 861-3075

Atty/Sec: CHM/mhn

NOTE: File in duplicate with 2 postcard receipts (PAT-103) & attachments.

532 Rec'd 5/17/00 20 APR 2001

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re PATENT APPLICATION OF

Confirmation No.: Unknown

VERKAMA

Group Art Unit: Unknown

Appln. No.: FILED HEREWITH

Examiner: Unknown

Filed: HEREWITH

Title: DIGITAL TELECOMMUNICATION SYSTEM

April 20, 2001

* * * * *

PRELIMINARY AMENDMENT

Hon. Commissioner of Patents
Washington, D.C. 20231

Sir:

Prior to initial examination on the merits, please amend the above-identified application as follows:

IN THE SPECIFICATION:

At the top of the first page, just under the title, insert

--This application is the National Phase of International Application PCT/FI99/00868 filed October 19, 1999 which designated the U.S. and that International Application was Published under PCT Article 21(2) in English.—

IN THE CLAIMS:

Please amend claims 1-14 as follows:

1. (Amended) A digital telecommunication system comprising:

a first centre configured to enable speech communication between a plurality of terminals, the first centre being associated with a calling terminal and including a first transcoder unit;

a second centre that is configured to enable speech communication between a plurality

of terminals, the second centre being associated with a called terminal and including a second transcoder unit,

wherein the first and second transcoder units each include speech codecs, the first centre is configured to perform handshaking with the second centre, the handshaking including indication of the speech codecs supported by the calling terminal, wherein at least one of the first and second centres is configured to choose the speech codec used by the calling and called terminals, and wherein at least one of the first and second centres is configured to establish call connections that bypass one or more of the transcoder units or to control the transcoder units to transmit encoded speech between the called and calling terminals without performing speech encoding operations so that speech is encoded and decoded only in the terminals.

2. (Amended) The telecommunication system of claim 1, wherein the telecommunication system is a mobile communication system in which the terminals include mobile stations, and the telecommunication system further comprises a mobile communication network and at least one of the first and second centres is a mobile switching centre.

3. (Amended) The telecommunication system of claim 2/ wherein:
the mobile switching centre includes a subscriber database configured to maintain subscriber data associated with a mobile subscriber, and the subscriber data includes information indicating the speech codecs supported by a mobile station associated with the mobile subscriber.

4. (Amended) The telecommunication system of claim 1, wherein the handshaking is

performed as outband signalling.

5. (Amended) The telecommunication system of claim 4, wherein the first and second centres are configured to perform the handshaking in association with a routing information inquiry issued in response to a determination that the called terminal is a mobile subscriber.

6. (Amended) The telecommunication system of claim 5, wherein:

the first centre is configured to send the routing information inquiry including information associated with the speech codecs supported by the calling terminal,

the second centre is configured to select a speech codec to be associated with the call connection which the calling and called terminals are configured to support, and

the second centre is configured to send information associated with the speech codec associated with the call connection in a reply message to the routing information inquiry.

7. (Amended) The telecommunication system of claim 6, wherein the routing information inquiry and reply message to the routing information inquiry are configured to pass via a home database of the called terminal.

8. (Amended) The telecommunication system of claim 4, wherein the first and second centres are configured to perform the handshaking in association with inter-MSC signalling.

9. (Amended) The telecommunication system of claim 8, wherein:

the first centre is configured to send a message requesting connection set-up, the message including information indicating the speech codecs supported by the calling terminal,

the second centre is configured to select a speech codec associated with the call connection which both the called and calling terminals are configured to support, and

the second centre is configured to send information associated with the codec associated with the call connection, in a reply message to the connection set-up message.

10. (Amended) The telecommunication system of claim 1, wherein, when required, at least one of the first and second centres is configured to notify the associated terminal of the speech codec it has to use as the result of the handshaking.

11. (Amended) The telecommunication system of claim 10, wherein at least one of the first and second centres is configured to notify the associated terminal of the speech codec to be used if it is not a default speech codec of the associated terminal.

12. (Amended) The telecommunication system of claim 1, wherein:
a pulse code modulated digital link exists between the first and second centres, and
the first and second centres are configured to control their respective transcoder units to adapt an encoded speech signal to one or more least significant bits of PCM samples without transcoding.

13. (Amended) The telecommunication system of claim 1, wherein:
the system is configured to support a packet-switched link between the first and second centres, and
the first and second centres are configured to connect a call connection that bypasses at least one of the transcoder units.

14. (Amended) A centre in a digital telecommunication network configured to connect a transcoder located in a transcoder unit to a call connection when required, wherein:

the centre is configured to perform handshaking with another centre associated with a called terminal, the handshaking including indication of speech codecs supported by the calling terminal associated with the centre, the centre also being configured to choose the speech codec used by the terminals, and

the centre is configured to connect a call connection that bypasses the transcoder unit or to control the transcoder unit to transmit the encoded speech without performing speech encoding operations in such a way that speech encoding and decoding are only performed in the calling or called terminal.

Please see the attached APPENDIX including the changes made to provide the above-claims.

Please add new claims 15-17.

15. (New) The telecommunication system of claim 8, wherein the inter-MSC signalling is ISUP signalling.

16. (New) The telecommunication system of claim 8, wherein the message requesting connection set-up is an IAM message according to ISUP signalling.

17. (New) The telecommunication system of claim 8, wherein the reply message to the connection set-up message is an ANM message according to ISUP signalling.

REMARKS

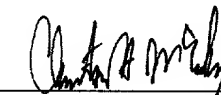
Consideration and allowance of the present application is respectfully requested. By this Amendment, claims 1-14 are amended to merely clarify the recited subject matter of the disclosed invention. Additionally, new claims 15-17 are added to more fully recite the claimed invention. No new matter is introduced by this Amendment, as new claims 15-17 are fully supported by the unamended claims and specification.

Attached hereto is a marked-up version of the changes made to the specification and claims by the current amendment. The attached Appendix is captioned "Version with markings to show changes made".

Applicant respectfully submits that the present application is in a condition for allowance and a Notice to that effect is earnestly solicited.

Respectfully submitted,
PILLSBURY WINTHROP LLP

By: _____



Christine H. McCarthy
Reg. No.: 41,844
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1100 New York Avenue, NW
Ninth Floor
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Enclosure: Appendix

APPENDIX

VERSION WITH MARKINGS TO SHOW CHANGES MADE

IN THE SPECIFICATION:

At the top of the first page, just under the title, insert

--This application is the National Phase of International Application PCT/FI99/00868
filed October 19, 1999 which designated the U.S. and that International Application was
Published under PCT Article 21(2) in English.--

IN THE CLAIMS:

Please amend claims 1-14 as follows:

1. (Amended) A digital telecommunication system [wherein terminals and a
telecommunication network comprise speech codecs, the] comprising:

a first centre configured to enable speech communication between a plurality of
terminals, the first centre being associated with a calling terminal and including a first
transcoder unit;

a second centre that is configured to enable speech communication between a plurality
of terminals, the second centre being associated with a called terminal and including a second
transcoder unit,

wherein the first and second transcoder units each include speech codecs [of the
telecommunication network being disposed in a transcoder unit, from which a centre in the
telecommunication network connects a transcoder for a speech connection, when required,
characterized in that]

, the first centre [of the calling terminal] is [arranged] configured to perform handshaking
with the second centre [of the called terminal], [said] the handshaking including [notification]
indication of the speech codecs supported by the calling terminal, wherein at least one of the

first and second centres is configured to choose the speech codec used by the calling and called terminals, and wherein at least one of the first and second centres is [are arranged] configured to establish call connections [past] that bypass one or more of the transcoder [unit] units or to control the transcoder units to [let] transmit [the] encoded speech [through] between the called and calling terminals without performing speech encoding operations so that speech is encoded and decoded only in the terminals.

2. (Amended) [A] The telecommunication system [as claimed in] of claim 1, **[characterized]** in that] wherein [said] the telecommunication system is a mobile communication system in which [said] the terminals [comprise] include mobile stations, [said] and the telecommunication system further [telecommunication network] comprises a mobile communication network and [said] at least one of the first and second centres is [centre of the telecommunication network comprises] a mobile switching centre.

3. (Amended) [A] The telecommunication system [as claimed in] of claim 2, **[characterized]** in that] wherein:
the mobile switching centre [comprises] includes a subscriber database [for maintaining] configured to maintain subscriber data [on] associated with a mobile subscriber [when the mobile station is located within the area of the mobile switching centre], and [said] the subscriber data [comprises] includes information [on] indicating the speech codecs supported by a mobile station associated with the mobile subscriber [subscriber's mobile station].

4. (Amended) [A] The telecommunication system [as claimed in any one] of [claims] claim 1 [to 3], **[characterized]** in that] wherein [said] the handshaking is [carried out] performed as outband signalling.

5. (Amended) [A] The telecommunication system [as claimed in] of claim 4,
[c h a r a c t e r i z e d in that] wherein the first and second [mobile switching] centres are
[arranged] configured to [carry out said] perform the handshaking in association with a
routing information inquiry issued in response to a determination that the called terminal is
[subscriber being] a mobile subscriber.

6. (Amended) [A] The telecommunication system [as claimed in] of claim 5,
[c h a r a c t e r i z e d in that] wherein:
the first [mobile switching] centre [of the calling subscriber] is [arranged] configured
to send [a] the routing information inquiry [comprising] including information [on] associated
with the speech codecs supported by the [mobile station] calling terminal,

the second [mobile switching] centre [of the called subscriber] is [arranged]
configured to select [for the call connection] a speech codec to be associated with the call
connection which the calling and called terminals [mobile stations of both the called and
calling subscribers] are configured to support, and

the second [mobile switching] centre [of the called subscriber] is [arranged]
configured to send information [on said] associated with the speech codec [, selected for]
associated with the call connection [,] in a reply message to the routing information inquiry.

7. (Amended) [A] The telecommunication system [as claimed in] of claim 6,
[c h a r a c t e r i z e d in that] wherein
[said] the routing information inquiry and reply message to the routing information
inquiry are [arranged] configured to pass via [the] a home database of the called [subscriber]
terminal.

8. (Amended) [A] The telecommunication system [as claimed in] of claim 4,
[c h a r a c t e r i z e d in that] wherein the [mobile switching] first and second centres are
[arranged] configured to [carry out said] perform the handshaking in association with inter-
MSC signalling[, such as ISUP signalling].

9. (Amended) [A] The telecommunication system [as claimed in] of claim 8,
[c h a r a c t e r i z e d in that] wherein:
the [mobile switching] first centre [of the calling subscriber] is [arranged] configured
to send a message requesting connection set-up[, such as an IAM message according to ISUP
signalling], the message [containing] including information [on] indicating the speech codecs
supported by the calling terminal[mobile station],

the [mobile switching] second centre [of the called subscriber] is [arranged]
configured to select [for the call connection] a speech codec associated with the call
connection which both the called and calling terminals [the mobile stations of both the called
and calling subscribers] are configured to support, and

the [mobile switching] second centre [of the called subscriber] is [arranged]
configured to send information [on said] associated with the codec [, selected for] associated
with the call connection, in a reply message to the connection set-up message[, such as in an
ANM message according to ISUP signalling].

10. (Amended) [A] The telecommunication system [as claimed in any one] of [the
preceding claims] claim 1, [c h a r a c t e r i z e d in that] wherein, when required, at least
one of the [mobile switching] first and second centres is [arranged] configured to notify the
[mobile station] associated terminal of the speech codec it has to use as the result of [said] the

handshaking.

11. (Amended) [A] The telecommunication system [as claimed in] of claim 10, **[c h a r a c t e r i z e d** in that] wherein
at least one of the first and second [mobile switching centre] centres is [arranged] configured to notify [the mobile station] the associated terminal of the speech codec to be used if it is not [the] a default speech codec of the [mobile station] associated terminal.

12. (Amended) [A] The telecommunication system [as claimed in any one] of [the preceding claims] claim 1, **[c h a r a c t e r i z e d** in that] wherein:

a pulse code modulated [(PCM)] digital link exists between the first and second [mobile switching] centres, and

the first and second [mobile switching] centres are [arranged] configured to control [the] their respective transcoder units [at the ends of said link] to adapt [the] an encoded speech signal to one or more least significant bits of PCM samples without transcoding.

13. (Amended) [A] The telecommunication system [as claimed in any one] of [claims] claim 1 [to 11], **[c h a r a c t e r i z e d** in that] wherein:

the system is configured to support a packet-switched link [exists] between the first and second [mobile switching] centres[, such as a network based on the ATM or IP technology], and

the first and second [mobile switching] centres are [arranged] configured to connect a call connection [past] that bypasses at least one of the transcoder [unit] units.

14. (Amended) A centre in a digital telecommunication network [, the centre being

arranged] configured to connect a transcoder located in a transcoder unit to a call connection when required, [**c h a r a c t e r i z e d** in that] wherein:

[said] the centre is [arranged] configured to perform handshaking with [the] another centre associated with [of] a called terminal, [said] the handshaking including indication [notification] of [the] speech codecs supported by the calling terminal associated with the centre, the centre also being configured to choose the speech codec used by the terminals, and

[said] the centre is [arranged] configured to connect a call connection [past] that bypasses the transcoder unit or to control the transcoder unit to [let] transmit the encoded speech [through] without performing speech encoding operations in such a way that speech encoding and decoding are only [carried out] performed in the calling or called terminal.

2/PRTS

DIGITAL TELECOMMUNICATION SYSTEM

The invention relates to a digital telecommunication system wherein terminals and a telecommunication network comprise speech codecs, the speech codecs of the telecommunication network being located in a transcoder unit, from which a centre in the telecommunication network connects a transcoder to a speech connection, when required.

In present digital mobile communication systems, speech and data are transferred entirely in digital form, resulting in a uniformly good quality of speech. As far as the mobile communication network is concerned, the most limited resource on a transmission path is the radio path between mobile stations and base stations. To make the bandwidth required by one radio connection on the radio path as narrow as possible, speech encoding is employed in speech transmission to allow significantly lower transmission rates than in a fixed telephone network (PSTN, Public Switched Telephone Network), for example. In this case a speech encoder and decoder have to exist both in the mobile station and on the side of the fixed mobile communication network. On the network side, speech encoding functions may be placed alternatively either in a base station or a mobile switching centre. Speech encoders and decoders are typically located far away from the base station, in what is known as remote transcoder units, speech encoding parameters being transferred in the network between a base station and the transcoder unit. Thus a transcoder unit is a part of the logical transmission path in a fixed mobile communication network from a base station to a mobile switching centre.

In mobile terminated (MT) or mobile originated (MO) speech calls, a transcoder is connected to the speech connection on the network side for encoding (downlink) speech signals destined to a mobile station and decoding (uplink) speech signals originating from a mobile station. This is necessary if one of the parties to a call is a mobile station and the other a subscriber in a public telephone network (PSTN), for example.

In the case of mobile-to-mobile calls (MMC), the above described connection of a transcoder to a call results in the mobile switching centre connecting two transcoder units in series to each MMC call, two speech encodings and decodings being performed on the call in the above described manner. This so-called tandem coding is a problem in mobile communication networks, since it weakens speech quality owing to the extra speech encoding and decoding. Consequently, methods for preventing tandem coding have

been developed in present digital mobile communication systems, for example the GSM system (Global System for Mobile communication). Methods of creating a tandem free function are based on signalling in a mobile communication network, the signalling comprising forwarding an indication to the transcoders upon set-up of an MMC call to the effect that they are to operate in a tandem coding prevention mode, whereby the transcoder does not at all encode or decode speech. Said signalling is transferred on a speech channel with speech parameters and other control information, i.e. as inband-signalling. In the tandem coding prevention mode, speech is encoded only in mobile stations and speech parameters are only transferred through the mobile communication network with slight changes from one base station via two tandem-connected transcoders to a second base station. This considerably improves the quality of speech as compared with a tandem coded MMC call.

In mobile communication networks, circuit-switched technology based on pulse code modulation (PCM) has been conventionally used in inter-MSC data transmission, i.e. PSTN or ISDN-based (Integrated Services Digital Network) network solutions. In this case, when a transcoder is in a tandem coding prevention mode, it combines control, synchronization and error correction information, for example, with speech parameters arriving from a mobile station via a base station, and adapts the data to PCM timeslots without transcoding. In mobile stations, encoded speech is adapted to a PCM channel such that one or more least significant bits of PCM samples constitutes a sub-channel into which lower-rate speech encoded by the mobile station is multiplexed. These PCM samples and their subchannels are transferred to the receiving transcoder which sends the speech parameters further to the receiving base station either as such or making slight changes indicated by the control information. Inter-MSC data transmission on a PCM channel is described in greater detail in the Applicant's previous Finnish patent application 960,590.

The above manner of arranging tandem coding prevention is a well working method in mobile communication systems in which transcoders are part of the transmission path of the mobile communication network, and in which PCM technology is used in inter-MSC data transmission. However, in future third generation mobile communication systems, the intention is not to place transcoders as part of the transmission path, but they are to be placed in what is known as a transcoder pool, in association with a mobile switching centre, for example. In this case the mobile switching centre connects a

transcoder to a call only if it is necessary, whereby the above manner of signalling a tandem coding prevention mode and adaptation of control information to speech parameters is not an advantageous way to implement a tandem free function. In third generation mobile communication systems, various alternative technologies are available for inter-MSC data transmission, including packet-switched connections not based on pulse code modulation. In this case it is not necessary to transmit inter-MSC signalling as part of a speech channel, which allows a simpler implementation of the tandem free function.

It is an object of the present invention to prevent tandem coding in calls between mobile stations by the use of simplified signalling better adaptable to new systems, in which the speech codec to be used is agreed upon between mobile switching centres.

The digital telecommunication system of the invention is characterized in that

the centre of the calling terminal is arranged to perform handshaking with the centre of the called terminal concerning the speech codec used by the terminals, and

the centres are arranged to establish call connections past the transcoder unit or to control the transcoder units to let the encoded speech through without speech encoding operations so that speech is encoded and decoded only in the terminals.

It is an essential idea of the invention that in a call between two mobile stations, the mobile switching centres of the calling and called mobile stations use mutual signalling to agree upon the speech codec to be used on a call connection. It is the idea of a preferred embodiment of the invention that, depending on the connection between the mobile switching centres, no transcoder is connected to the call connection. It is the idea of another preferred embodiment of the invention that said signalling is what is known as outband signalling.

It is an advantage of the invention that the signalling of the invention simplifies implementation of the tandem free function, as transcoders are no longer automatically part of the transmission path. The signalling of the invention provides a common starting point for inter-MSC transmission of a call between two mobile stations irrespective of what kind of a connection is in use between the mobile switching centres. It is a further advantage of a preferred embodiment of the invention that, since, depending on the connection

between the mobile switching centres, no transcoder is connected to the call connection, speech parameters do not have to be adapted to PCM frames as is the case in present transcoders. Neither do the transcoders necessarily have to support a speech codec to be used in calls between two mobile stations, and consequently mobile station-specific speech codecs can be rapidly taken into use in new systems. Still another advantage of the invention is that present network elements and signalling architecture in a mobile communication network can be used. New signalling messages, for example, do not have to be created for implementing the invention, but the invention can be implemented by modifying the contents of existing messages.

In the following the invention will be described in greater detail with reference to the accompanying drawings, in which

Figure 1 is a simplified block diagram of an architectural model of a third generation mobile communication system,

Figure 2 shows a transport cell according to a packet-switched transmission method which can be utilized in a preferred embodiment of the invention,

Figure 3 shows an adaptation protocol function of a packet-switched transmission method which can be utilized in a preferred embodiment of the invention,

Figure 4 shows protocol layers of a packet-switched transmission method which can be utilized in a second preferred embodiment of the invention, and

Figure 5 shows call set-up signalling according to some preferred embodiments of the invention.

In this context, the term speech codec, or simply codec, refers to a functional entity which serves to encode or decode speech into a form required by a mobile communication system.

Figure 1 is a simplified block diagram of an architectural model of a third generation mobile communication system. The design of core network solutions in third generation mobile communication systems is based on the present European digital mobile communication system GSM. This allows the use of present core network solutions also in the future almost as such, and only changes required by new functions and services will be made. This provides considerable savings, since the expensive core networks do not have to be completely rebuilt. This is why reference is made in the examples of the

present description, when applicable, to the present GSM system, since, for the most part, the principals of the signalling inside the core network will remain the same.

In Figure 1, a mobile station (MS) communicates with a wideband mobile services switching centre (WMSC) via a radio access network (RAN). The radio network RAN comprises a base station system (not shown) comprising base transceiver stations (BTS) and radio network controllers (RNC), and signalling between them, but as far as the invention is concerned, the radio network may also be structurally different. Wideband CDMA technology, i.e. WCDMA technology, is used at the radio interface between the mobile station MS and the radio network RAN. However, the radio technology used is not relevant to the invention, and consequently the invention can also be used in systems applying other technologies. The radio network RAN communicates with the mobile switching centre WMSC over a radio interface Iu, for whose standards the ETSI (European Telecommunications Standards Institute) is currently drawing up recommendations. The mobile switching centres WMSC also have visitor location registers (VLR) and transcoder units (TCU). The mobile switching centres WMSC signal to a home location register (HLR) information on the user of the mobile station, i.e. the subscriber, concerning access rights, functions and charging, for example. MAP (Mobile Application Part) is the abbreviation generally used for referring to this signalling and it is described in greater detail in GSM recommendation 09.02 *Mobile Application Part (MAP)*. Said subscriber data is also stored in the visitor location register VLR when a mobile station MS visits the area of the corresponding mobile switching centre WMSC.

In a preferred embodiment of the present invention, mobile switching centres agree by mutual handshaking signalling upon the speech codec to be used in an MMC call between two mobile stations, MS1 and MS2, whereupon, depending on the connection between the mobile switching centres, the call is either connected past the transcoder unit or the transcoder unit is controlled to let the call pass through without speech encoding functions on the mobile communication network side in such a manner that speech is encoded and decoded only in the mobile stations MS1 and MS2. According to the invention, this is achieved by indicating the speech codecs supported by the mobile station MS1 of subscriber A to the mobile switching centre WMSC(A) of subscriber A. The mobile switching centre WMSC(A) stores this information

in the visitor location register VLR(A), attaches said information as part of a routing information inquiry to be sent to the home location register HLR, and the home location register HLR relays the information further to the mobile switching centre WMSC(B) of subscriber B. Subscribers A and B may also be attached to the same mobile switching centre, in which case the routing information inquiry does not have to be sent via the home location register HLR, but it can be made via the visitor location register VLR in association with the mobile switching centre WMSC. The speech codecs supported by the mobile station MS2 of subscriber B are also indicated to the mobile switching centre WMSC(B) of subscriber B, and the mobile switching centre WMSC(B) stores this information in the visitor location register VLR(B). The mobile switching centre WMSC(B) of subscriber B selects a codec suitable for both mobile stations, MS1 and MS2, informs the mobile switching centre WMSC(A) of subscriber A, and stores the information on the codec to be used in its database VLR(B).

In a preferred embodiment of the present invention, an MMC call between two mobile stations MS1 and MS2 can be so switched that no transcoder at all is connected to the connection. This is carried out as follows: after the above described signalling, the mobile switching centres having agreed upon the speech codec to be used on the call connection, the mobile switching centre WMSC(A) checks the transmission technology the connection uses between the mobile switching centres WMSC(A) and WMSC(B). If pulse code modulation is not used on said connection, i.e. the connection is packet-switched, for example, then, in response to this, the mobile switching centre WMSC(A) connects no transcoder to the connection. Alternatively, the connection between the mobile switching centres WMSC(A) and WMSC(B) may be a PCM-switched PSTN or ISDN connection. In this case the mobile switching centre WMSC(A) controls the transcoder unit TCU(A), in a manner known per se, to switch the call connection via the transcoder without speech encoding functions in such a way that speech is encoded and decoded only in the mobile stations MS1 and MS2.

Third generation mobile stations use various speech 1codecs, and in MMC calls, to which no transcoder is connected in the above manner, it is essential that mobile stations use the same kind of speech codec. According to a preferred embodiment of the invention, the speech codec to be used is indicated, when required, to both mobile stations before the call is switched. A

default codec to be used by the mobile stations MS1 and MS2, unless otherwise notified, is preferably defined. Similarly, the visitor location registers VLR(A) and VLR(B) comprise information on the default speech codecs. Should the above handshaking signalling result in the use of another speech
5 codec on the call connection than is the default set for the mobile stations MS1 or MS2, information on this is forwarded to the mobile switching centres WMSC(A) and WMSC(B). Finally, when the call is set up, the mobile switching centres WMSC(A) and WMSC(B) inform the mobile stations MS1 and MS2, respectively, which codec to use, should it not be the default codec.

10 In accordance with a second preferred embodiment of the invention, handshaking signalling concerning the speech codec to be used is carried out as part of physical call set-up. In this case the speech codec to be used is notified to the mobile switching centre WMSC(A) in a reply message to a call set-up message, whereupon the mobile switching centres WMSC(A)
15 and WMSC(B) inform, when required, the mobile stations MS1 and MS2 about the codec to be used, and control the transcoder units TCU(A) and TCU(B) in a manner required by the transmission connection, as was described above.

In third generation mobile communication networks, inter-WMSC traffic is designed to be carried out by packet-switched connections, when
20 possible. In other words, it can be preferably carried out by means of wide-band ATM network technique (Asynchronous Transfer Mode), for example. ATM is a general-purpose transfer mode which combines the advantages of circuit-switched and packet-switched data transmission. ATM is based on cell-switched data transmission, the data to be transmitted being split into bits
25 having a given length, i.e. cells. Telecommunication applications which require constant capacity or delay and which have conventionally used a circuit-switch connection, are prioritized in filling the cells. Applications not requiring constant capacity or delay, transmit their data in the remaining cells in the same way as on a packet-switched connection.

30 An ATM cell comprises 53 bytes, of which 48 bytes are payload and 5 bytes are reserved for header data. Figure 2 shows an ATM cell and its header fields. A GFC field (Generic Flow Control) is used in connection flow control. A virtual path identifier (VPI) indicates to the ATM network switches the route of the cell in the network, cells having the same VPI value being
35 always transmitted to the same address. A virtual channel identifier (VCI) operates like the VPI, and both VPI and VCI values are used in defining a logical

channel, allowing the simultaneous connection of whole channel groups to the backbone network. Hereby the VPI between two functional points can be agreed upon among service providers, and yet the service user is able to define the VCI values. The type of payload is defined in a PT field (Payload Type). A CLP field (Cell Loss Priority) allows traffic to be divided into two classes, resulting in the cells whose CLP bit = 1 being destroyed first when the network gets congested. An HEC field (Header Error Correction) is used to ascertain the correctness of header bits.

ATM technique can be utilized in various applications, and therefore the need has arisen to define adaptation protocols (AAL, ATM Adaptation Layer) for different application types. Figure 3 shows an AAL function in which a data packet originating from a mobile switching centre, for example, is split in the ATM adaptation function into 48-byte cells, which are further applied to ATM circuits, which attach a five-byte header to the cells. In the physical access layer these cells are further set into an SDH form (Synchronous Digital Hierarchy), which specifies in optical fibre-based transmission systems how data streams at different rates are transmitted in the backbone network. The ATM backbone network is composed of ATM switches, which are linked together by high-rate connections, usually optical fibres, and to which local networks, mobile switching centres, telephone exchanges or video devices, for example, can be further connected. In present ATM networks, the transfer rate may vary, depending on the connection, between 64 kbps and 622 Mbps, but in the future several Gbps will be reached. As to a more precise description of the ATM technique, reference is made to '*Asynchronous Transfer Mode: Atm Architecture and Implementation*'; J. Martin, K. Chapman, J. Leben; Prentice Hall, USA; ISBN: 0135679184.

During the last few years, the use of the Internet has grown exponentially and become more versatile, new services and options being continuously developed. The TCP/IP protocol (Transmission Control Protocol/Internet Protocol) acts as the data transmission protocol in the Internet, the special advantage being its independence of different device or software architectures, which makes it the most generally used network protocol in the world, especially in local networks. In Internet-based networks, the IP protocol is the actual network protocol which serves to route an addressed IP message from a source station to a destination station. A transport protocol, either TCP or UDP (User Datagram Protocol), is run above the IP network protocol. The

transport protocol attends to the transfer of data packets from a source port to a destination port. The TCP offers reliable connections to applications, i.e. the TCP splits the data from the applications into IP packets, sees to it that the data arrives intact and in the right order, resends lost or damaged data packets and also attends to flow control. The UDP, in turn, is a lighter transport protocol than the TCP and does not answer for the arrival or correctness of data packets. This makes the UDP an unreliable transport protocol, which leaves error and correctness checks to the application program, but is better suited to services requiring real-time performance.

The generality of Internet-based networks and the inexpensive, in local networks even free data transmission, they offer, have aroused great interest in switching also voice calls via IP networks. This would also allow inter-MSC data to be transmitted by means of IP networks. The device and system solutions thus far developed for transmitting conventionally circuit-switched voice calls in a packet-switched IP network are rather unreliable and incompatible. To make Internet call systems compatible, a standard (VoIP, Voice over IP) is being created, for example for determining the compatibility of devices, quality of service, and routing calls in IP networks.

Figure 4 shows a VoIP standard recommendation for the protocol stack in Internet call systems. Either the TCP or the UDP is run above the IP network protocol, depending on the application. At the next layer, an H.323 protocol stack is placed; a standard defined by the ITU (International Telecommunication Union) for packing voice and video image used in video conference programs and for controlling calls. The H.323 is used for call set-up and adaptation negotiations, and for reserving a connection required by real-time speech in an IP network. Call control and functions and services associated therewith, such as choice of transfer protocol, optional speech encoding, voice activity detection (VAD) and DTMF functions, are carried out in a CMAS unit (Call Management Agent System) comprising CMA framing and agents for each function (Basic Agents). The CMAS utilizes the LDAP (Lightweight Directory Access Protocol) for dealing with the name service in telecommunications between different types of networks and file servers without the transport protocol having to deal with it. An external telephone network, for example a mobile telephone network, can be linked to the VoIP system by means of an H.323 gateway server (not shown). In fact, a mobile telephone operator is able to best utilize the VoIP system in his own local or wide area network

(LAN/WAN), allowing the operator to manage traffic both in the network and in the H.323 gateway servers.

Data transmission protocols based on the ATM and IP technologies are presented herein by way of example as data transmission technologies advantageous to the implementation of the invention. They use packet-switched data transmission, i.e. data frames are not adapted to PCM timeslots. This provides the advantage that, as no adaptation to PCM frames is required, a call can be set up completely without transcoders. Inter-MSC handshaking signalling can also be carried out as outband signalling, allowing the handshaking signalling to be carried out separately from call set-up, for example directly in inter-MSC connection set-up. It is obvious that the mobile communication system of the invention can be implemented by the use of any corresponding packet-switched data transmission technology, e.g. by means of xDSL technology (Digital Subscriber Line).

In the following, a preferred embodiment of the invention will be described with reference to Figure 5. Figure 5 only shows the relaying of messages relevant to the implementation of the invention in a mobile communication system. Consequently, between the messages described, messages may be relayed that are not essential to the implementation of the invention. The speech codecs supported by the mobile station MS1 of subscriber A are indicated to the mobile switching centre WMSC(A). This may preferably take place during call set-up signalling as the mobile station MS1 requests connection set-up of the mobile communication network, whereby the mobile switching centre WMSC(A) can store the data on the speech codecs supported by the mobile station MS1 in the visitor location register VLR(A). For data transmission a classmark identifier can also be used, which is known for example from the GSM system and comprises data on the properties of a mobile station and which the mobile station sends to the network when requested or when the mobile station wants to change classmark classes. Similarly, the speech codecs supported by the mobile station MS2 of subscriber B are indicated to the mobile switching centre WMSC(B). Relaying call set-up signalling and classmark identifiers is described in greater detail in GSM recommendation 04.08 *Mobile radio interface layer 3 specification*.

When subscriber A initiates call set-up, the mobile station MS1 sends via the radio network RAN to the mobile switching centre WMSC(A) a call setup message, on the basis of which the mobile switching centre

WMSC(A) identifies the called subscriber B as a mobile station. Subscriber B is identified on the basis of a numerical analysis, the identification being known per se from optimal call routing (OR), for example. In accordance with Figure 5, the mobile switching centre WMSC(A) receives a CM_SER_REQ message (Connection Management_Service_Request), for example, as a sign of initiation of call set-up. In order for the call to be able to be routed to subscriber B via the right mobile switching centre WMSC(B), the mobile switching centre WMSC(A) sends to the home location register HLR a routing information inquiry MAP_SRI (MAP_Send_Routing_Information), to which is attached information on the speech codecs supported by the mobile station MS1, preferably in the preference order of the mobile station MS1. The preference order serves to always use the default speech codecs of mobile stations, as far as is possible. The home location register HLR attaches this information further as part of a roaming number inquiry sent to the visitor location register VLR(B) of the mobile switching centre WMSC(B), MAP_PRN (MAP_Provide_Roaming_Number). The mobile switching centre WMSC(B) selects from the speech codecs informed the one that is suitable for the mobile station MS2, making the selection preferably in the preference order given by the mobile station MS1. Information on the speech codec selected is stored in the visitor location register VLR(B) and attached to a roaming number reply MAP_PRN_ack sent to the home location register HLR. The home location register HLR further attaches the information to a reply message to the routing information inquiry, MAP_SRI_ack, which is sent to the mobile switching centre WMSC(A) which stores the information in the visitor location register VLR(A).

As call set-up progresses, the mobile switching centre WMSC(B) sends to the visitor location register VLR(B) an inquiry of necessary authentication and encryption information. The corresponding inquiry for subscriber A is already made at the initial stage of call set-up in a message MAP_PAR (MAP_Process_Access_Request). To initiate actual call switching, both visitor location registers VLR(A) and VLR(B) issue to the mobile switching centres WMSC(A) and WMSC(B), respectively, a command MAP_COMPLETE_CALL, to which information on the speech codec selected for that call connection is attached. If the speech codec selected for the call connection is not the default speech codec of mobile stations MS1 or MS2, the mobile switching centres transmit information on the selected speech codec further to the mobile stations. Then, in the MO section of the call, the WMSC(A) indicates the in-

formation to the MS1 in a message CALL_PROC and, similarly, in the MT section of the call, the WMSC(B) indicates the information to the MS2 in a SETUP message. In response to this, both mobile stations MS1 and MS2 connect the same speech codec to the call.

5 Now, if packet-switched ATM technology, for example, is used on the connection between the mobile switching centres WMSC(A) and WMSC(B) instead of data transmission based on circuit-switched PCM technology, no transcoder at all is connected to the connection, but the speech frames encoded by the mobile station MS1 by means of the above AAL function of the ATM, suitable for the mobile switching centre, are placed in ATM cells. Similarly, when the VoIP technology is used, speech frames are placed by means of the H.323 gateway server into H.323 frames complying with the VoIP standard. In this case, as far as the fixed mobile communication network is concerned, speech frames are transmitted in the exact speech frame form encoded by the mobile station. If again the inter-MSC connection utilizes the PSTN or ISDN technology, the mobile switching centres connect transcoders to the connection and control these to adapt the speech frames encoded by the mobile station to the PCM form required by the PSTN and ISDN technologies, however, without transcoding. In this case the adaptation function carried out by the transcoders corresponds to the tandem free function of the known GSM technology.

 A second preferred embodiment of the invention can be implemented in a mobile communication system allowing direct signalling on an inter-MSC connection. One such signalling model is what is known as ISUP signalling (ISDN User Part), usable in inter-MSC signalling. ISUP signalling is described in greater detail in the ITU standard recommendations Q.721-Q.764. In accordance with Figure 5, in inter-MSC signalling three ISUP messages are used: IAM (Initial Address Message), ACM (Address Complete Message) and ANM (Answer Message). In accordance with the invention, the speech codecs supported by subscriber A are then notified to the mobile switching centre WMSC(B) of subscriber B in an IAM message, allowing non-defined spare values of the IAM message to be advantageously utilized. The mobile switching centre WMSC(B) of subscriber B sends an ACM message to the mobile switching centre WMSC(A) after a SETUP message sent to the mobile station MS2. The mobile switching centre WMSC(B) and the mobile station MS2 set up the connection by messages CONN (Connect) and

CONN_ack. The mobile switching centre WMSC(B) selects the speech codec in the same way as was described above and attaches information on the speech codec selected as part of an ANM message sent to the mobile switching centre WMSC(A).

5 In the present embodiment of the invention, information on the speech codec selected is not transferred to the mobile switching centre WMSC(A) of subscriber A until the physical transmission path has been set up. Consequently, in an MMC call between two mobile stations MS1 and MS2, the transcoder units in the mobile switching centres are not controlled to
10 switch the call past the transcoder unit or to control the transcoder unit to let the call through without speech encoding operations until after connection set-up. In other respects than the handshaking signalling of the speech codecs and the control of the transcoder units, this embodiment of the invention can be implemented in the same way as was described above. The implementa-
15 tion of this embodiment of the invention also allows the use of any other inter-MSC signalling, such as TUP signalling (Telephone User Part).

The invention and the signalling associated therewith have been described herein according to potential embodiments of the invention and only to the degree that the description of the signalling is relevant to the imple-
20 mentation of the invention. As to a more precise description of signalling, particularly as to functions under malfunction, reference is made to the GSM recommendation 09.02 *Mobile Application Part (MAP)*, Chapter 18, 'Call Handling Procedures' (v. 4.18.0).

Even though the invention has been described herein with mobile
25 communication systems as the basis, the principles of the invention can be implemented in any corresponding telecommunication system in which centres perform handshaking concerning speech codecs used by terminals. The invention is particularly applicable in mobile communication systems, since said environment uses a plurality of different terminals in which a plurality of differ-
30 ent speech encoding methods are used, the interfaces between the terminals and the network being accurately standardized.

The figures and the related specification are only intended to illustrate the present invention. It is obvious to a person skilled in the art that the details of the invention may be implemented in a variety of ways within the
35 scope of the attached claims.

Claims

1. A digital telecommunication system wherein terminals and a tele-
communication network comprise speech codecs, the speech codecs of the
telecommunication network being disposed in a transcoder unit, from which a
5 centre in the telecommunication network connects a transcoder for a speech
connection, when required, **characterized** in that

the centre of the calling terminal is arranged to perform handshak-
ing with the centre of the called terminal, said handshaking including notifica-
tion of the speech codecs supported by the calling terminal, to choose the
10 speech codec used by the terminals, and

the centres are arranged to establish call connections past the
transcoder unit or to control the transcoder units to let the encoded speech
through without speech encoding operations so that speech is encoded and
decoded only in the terminals.

15 2. A telecommunication system as claimed in claim 1, **character-
terized** in that

said telecommunication system is a mobile communication system
in which said terminals comprise mobile stations, said telecommunication net-
work comprises a mobile communication network and said centre of the tele-
20 communication network comprises a mobile switching centre.

3. A telecommunication system as claimed in claim 2, **character-
terized** in that

the mobile switching centre comprises a subscriber database for
maintaining subscriber data on a mobile subscriber when the mobile station is
25 located within the area of the mobile switching centre, and

said subscriber data comprises information on the speech codecs
supported by the subscriber's mobile station.

4. A telecommunication system as claimed in any one of claims 1 to
3, **characterized** in that

30 said handshaking is carried out as outband signalling.

5. A telecommunication system as claimed in claim 4, **character-
terized** in that

the mobile switching centres are arranged to carry out said hand-
shaking in association with a routing information inquiry in response to the
35 called subscriber being a mobile subscriber.

6. A telecommunication system as claimed in claim 5, **character-**

terized in that

the mobile switching centre of the calling subscriber is arranged to send a routing information inquiry comprising information on the speech codecs supported by the mobile station,

5 the mobile switching centre of the called subscriber is arranged to select for the call connection a speech codec which the mobile stations of both the called and calling subscribers support, and

the mobile switching centre of the called subscriber is arranged to send information on said speech codec, selected for the call connection, in a
10 reply message to the routing information inquiry.

7. A telecommunication system as claimed in claim 6, **characterized** in that

said routing information inquiry and reply message to the routing information inquiry are arranged to pass via the home database of the called
15 subscriber.

8. A telecommunication system as claimed in claim 4, **characterized** in that

the mobile switching centres are arranged to carry out said handshaking in association with inter-MSC signalling, such as ISUP signalling.

20 9. A telecommunication system as claimed in claim 8, **characterized** in that

the mobile switching centre of the calling subscriber is arranged to send a message requesting connection set-up, such as an IAM message according to ISUP signalling, the message containing information on the speech
25 codecs supported by the mobile station,

the mobile switching centre of the called subscriber is arranged to select for the call connection a speech codec which the mobile stations of both the called and calling subscribers support, and

the mobile switching centre of the called subscriber is arranged to
30 send information on said codec, selected for the call connection, in a reply message to the connection set-up message, such as in an ANM message according to ISUP signalling.

10. A telecommunication system as claimed in any one of the preceding claims, **characterized** in that

35 when required, at least one of the mobile switching centres is arranged to notify the mobile station of the speech codec it has to use as the re-

sult of said handshaking.

11. A telecommunication system as claimed in claim 10, **characterized** in that

the mobile switching centre is arranged to notify the mobile station
5 of the speech codec to be used if it is not the default speech codec of the mobile station.

12. A telecommunication system as claimed in any one of the preceding claims, **characterized** in that

a pulse code modulated (PCM) digital link exists between the mobile switching centres, and

the mobile switching centres are arranged to control the transcoder units at the ends of said link to adapt the encoded speech signal to one or more least significant bits of PCM samples without transcoding.

13. A telecommunication system as claimed in any one of claims 1 to 11, **characterized** in that

a packet-switched link exists between the mobile switching centres, such as a network based on the ATM or IP technology, and

the mobile switching centres are arranged to connect a call connection past the transcoder unit.

14. A centre in a digital telecommunication network, the centre being arranged to connect a transcoder located in a transcoder unit to a call connection when required, **characterized** in that

said centre is arranged to perform handshaking with the centre of a called terminal, said handshaking including notification of the speech codecs supported by the calling terminal, to choose the speech codec used by the terminals, and

said centre is arranged to connect a call connection past the transcoder unit or to control the transcoder unit to let the encoded speech through without speech encoding operations in such a way that speech encoding and decoding are only carried out in the terminal.

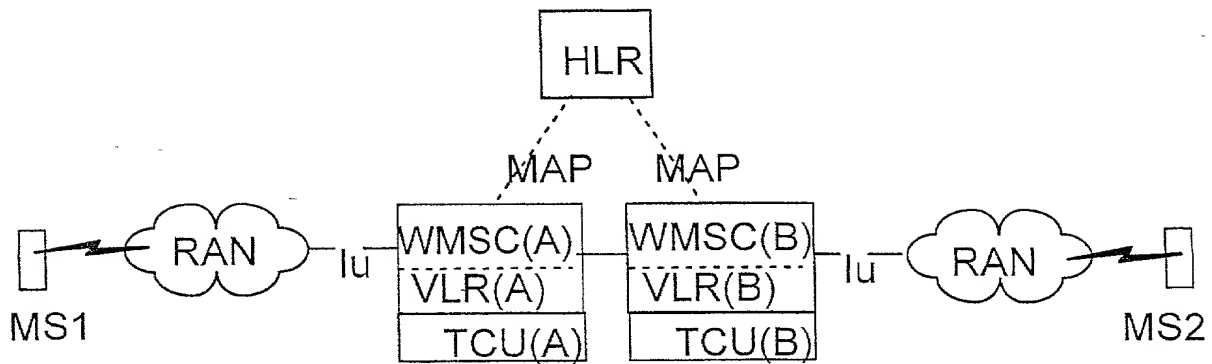


FIG. 1

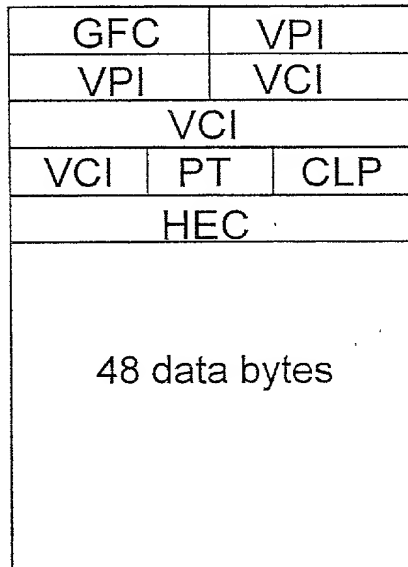


FIG. 2

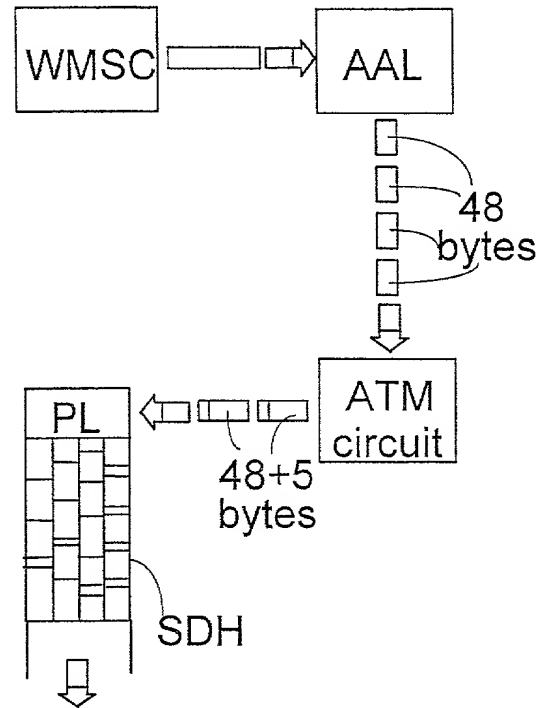


FIG. 3

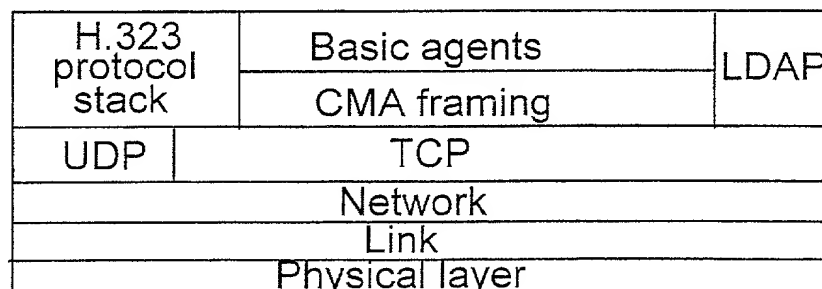


FIG. 4

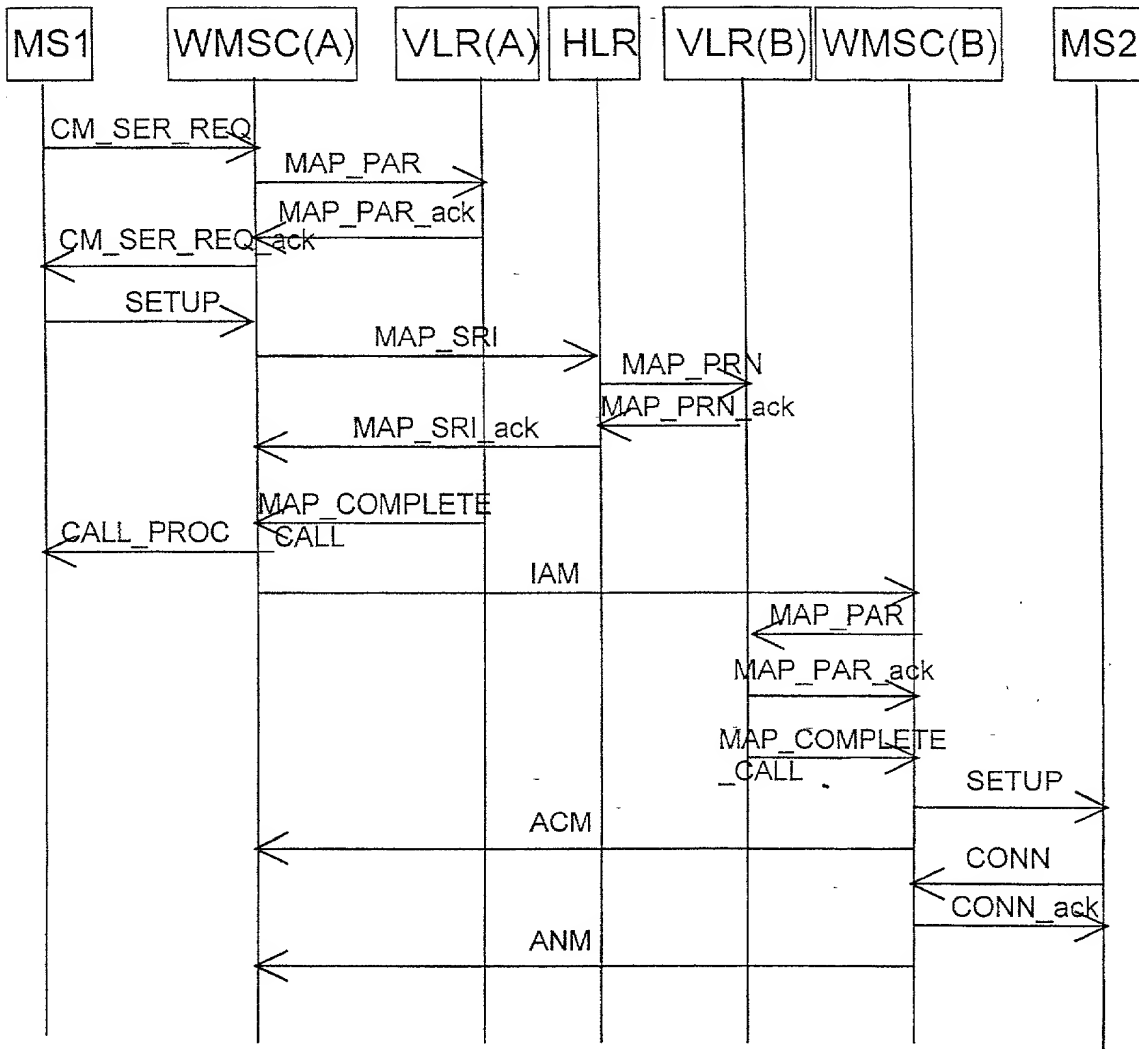


FIG. 5

FOR UTILITY/DESIGN
CIP/PCT NATIONAL/PLANT
ORIGINAL/SUBSTITUTE/SUPPLEMENTAL
DECLARATIONS

RULE 63 (37 C.F.R. 1.63)
DECLARATION AND POWER OF ATTORNEY
FOR PATENT APPLICATION

PM & S
FORM

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

As a below named inventor, I hereby declare that my residence, post office address and citizenship are as stated below next to my name, and I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the INVENTION ENTITLED
DIGITAL TELECOMMUNICATION SYSTEM

the specification of which (CHECK applicable BOX(ES))

X A. ☐ is attached hereto

BOX(ES) B. ☐ was filed on

as U.S. Application No. /

→ C. X was filed as PCT International Application No. PCT /F199 /00868 on 19 October 1999

and (if applicable to U.S. or PCT application) was amended on 5 December 2000

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims, as amended by any amendment referred to above. I acknowledge the duty to disclose all information known to me to be material to patentability as defined in 37 C.F.R. 1.56. Except as noted below, I hereby claim foreign priority benefits under 35 U.S.C. 119(a)-(d) or 365(b) of any foreign application(s) for patent or inventor's certificate, or 365(a) of any PCT International Application which designated at least one other country than the United States, listed below and have also identified below any foreign application for patent or inventor's certificate, or PCT International Application, filed by me or my assignee disclosing the subject matter claimed in this application and having a filing date (1) before that of the application on which priority is claimed, or (2) if no priority claimed, before the filing date of this application

PRIOR FOREIGN APPLICATION(S)

Number
982283

Country
Finland

Day/MONTH/Year Filed
21 October 1998

Date first Laid-
open or Published

Date Patented
or Granted

Priority NOT Claimed

If more prior foreign applications, X box at bottom and continue on attached page.

Except as noted below, I hereby claim domestic priority benefit under 35 U.S.C. 119(e) or 120 and/or 365(c) of the indicated United States applications listed below and PCT international applications listed above or below and, if this is a continuation-in-part (CIP) application, insofar as the subject matter disclosed and claimed in this application is in addition to that disclosed in such prior applications, I acknowledge the duty to disclose all information known to me to be material to patentability as defined in 37 C.F.R. 1.56 which became available between the filing date of each such prior application and the national or PCT international filing date of this application

PRIOR U.S. PROVISIONAL, NONPROVISIONAL AND/OR PCT APPLICATION(S)

Application No. (series code/serial no.)

Day/MONTH/Year Filed

Status

Priority NOT Claimed

pending, abandoned, patented

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

And I hereby appoint Pillsbury Winthrop LLP, Intellectual Property Group, 1100 New York Avenue, N.W., Ninth Floor, East Tower, Washington, D.C. 20005-3918, telephone number (202) 861-3000 (to whom all communications are to be directed), and the below-named persons (of the same address) individually and collectively my attorneys to prosecute this application and to transact all business in the Patent and Trademark Office connected therewith and with the resulting patent, and I hereby authorize them to delete names/numbers below of persons no longer with their firm and to act and rely on instructions from and communicate directly with the person/assignee/attorney/firm/ organization who/which first sends/sent this case to them and by whom/which I hereby declare that I have consented after full disclosure to be represented unless/until I instruct the above firm and/or a below attorney in writing to the contrary

Paul N. Kokulis	16773	Paul E. White, Jr.	32011	Stephen C. Glazier	31361	William P. Atkins	38821
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G. Lloyd Knight	17698	Kendrew H. Colton	30368	Richard H. Zaitlen	27248	Robin L. Teskin	35030
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Mailing Address		
(include Zip Code)		

"X" box ☐ FOR ADDITIONAL INVENTORS, and proceed on the attached page to list each additional inventor.

☐ See additional foreign priorities on attached page (incorporated herein by reference).

Atty. Dkt. No. PM